(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 987 894 A2

(12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
22.03.2000 Bulletin 2000/12

(51) Int Cl.7: H04N 7/24

(21) Application number: 99304426.2

(22) Date of filing: 08.06.1999

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(30) Priority: 07.09.1998 SG 9803578

(71) Applicant: VICTOR COMPANY OF JAPAN, LTD.
Yokohama-Shi, Kanagawa-Ken (JP)

(72) Inventor: XU, Jinghong
Singapore 760247 (SG)

(74) Representative: Crawford, Andrew Birkby et al
A.A. Thornton & Co.
235 High Holborn
London WC1V 7LE (GB)

(54) A dejittering and clock recovery technique for real-time audio/visual network applications

(57) In a real-time audio/visual system in which A/V data is conveyed over a jitter-introducing network, dejittering and clock recovery processes can be achieved without requiring a Phase Locked Loop (PLL). At the server, audio/video streams are encoded into transport packets before being sent out. At the client, the dejittering process is achieved by a dejittering buffer using the embedded timestamps in the transport packets and a client decoding clock. The delay variations of data arriving are removed after the client buffering process. At the scheduled time, each data packet is shifted to a synchronizing buffer and then fed to the A/V decoder according to the speed of A/V stream. The clock synchronization between client and server is achieved by a synchronizing buffer whose half-size position is taken as the reference. By monitoring the movement of the buffer fill position over a given period, the drift rate of clock unsynchronization between client and server can be derived and, therefore, the client's clock can be adjusted to synchronize with the server's clock based on the derived drift.

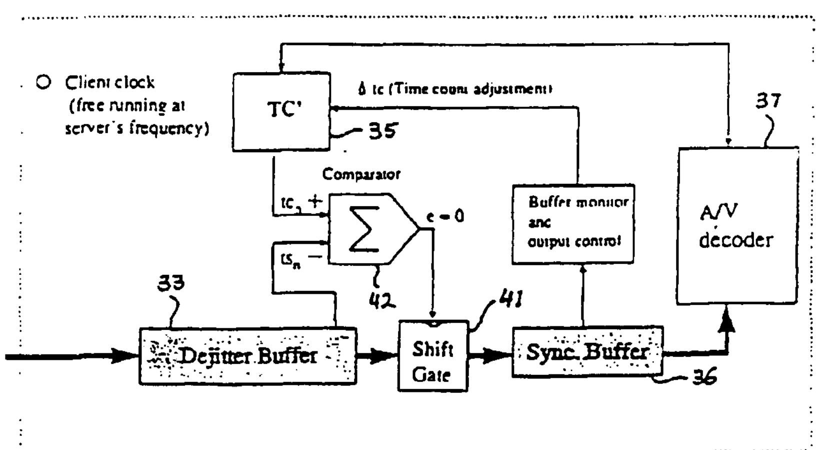

Figure 2

EP 0 987 894 A2

[0006] It is an object of the present invention, to provide a technique for dejittering and clock recovery for use in the client application of a networked real-time audio/visual service system. It is desirable to be able to implement embodiments of the invention to synchronize the clients to a server in a jitter introducing network environment without employing additional devices or a special decoder.

Summary of the Invention

[0007] In accordance with the present invention, there is provided a method for clock variation compensation in a real-time audio/visual system in which encoded A/V data in a plurality of data packets are delivered to at least one client over a network from a server at a substantially constant bit rate, said plurality of data packets including data packets containing time stamp data the method comprising the steps of:

receiving and buffering the data packets in a first buffer at a client;
passing selected data packets from the first buffer to a second buffer at scheduled times based on a comparison between a local clock of the client and timestamp data corresponding to the selected data packets; and
passing the data in the second buffer to a data decoder of the client.

[0008] Preferably the method includes monitoring the fullness of the second buffer to derive a drift rate, and adjusting the client local clock based on the derived drift rate.

[0009] The present invention also provides a method for clock variation compensation in a real-time audio/visual system in which encoded A/V data in a plurality of data packets are delivered to at least one client over a network from a server at a substantially constant bit rate, said plurality of data packets including data packets containing timestamp data, the method comprising the steps of:

receiving and buffering the data packets in a dejittering buffer at a client;
passing selected data packets from the dejittering buffer to a data decoder of the client at scheduled times based on a comparison between a decoding system clock of the client and timestamp data corresponding to the selected data packets.

[0010] Preferably the method includes monitoring the fullness of the dejittering buffer to derive a clock drift rate, and adjusting the decoding system clock based on the derived drift rate and a network jitter component.

[0011] The present invention further provides a real-time audio/visual system coupled to receive data packets over a network for A/V decoding by an A/V decoder, the system including a clock variation compensation system comprising:

a dejitter buffer for receiving and storing packets of data from the network;
a synchronization buffer for feeding data for decoding to the A/V decoder;
a decoder system clock; and
a buffer data flow controller for controlling the passing of selected data packets from the dejitter buffer to the synchronization buffer in accordance with a comparison of a first signal derived from the decoder system clock and a second signal derived from a timestamp from the selected data packets.

[0012] The technique summarized below is a "software PLL-like" method to address the jitter and time synchronization for a client decoding system. It employs the RTP (Real-Time Transport Protocol) as the transport service and receiver buffering to achieve real-time A/V playback.

[0013] The dejittering process can be achieved by a dejittering buffer using the embedded timestamp values in the transport packets and client RTP clock (which runs at the same frequency as the A/V decoder's clock). The delay variations of data arriving are removed after the client buffering process, the data packet is shifted to a synchronizing buffer at the scheduled time of server encoder, then feed to the A/V decoder according to the time reference of A/V encoder. The clock synchronization (recovery) can be achieved by a synchronizing buffer based on a reference fill position and the movement of the fill position in the buffer (packet index oriented). By monitoring the fullness of the buffer over a given period, the drift rate of clock unsynchronization between client and server can be derived.

Brief Description of the Drawings

[0014] The invention is described in greater detail hereinafter, by way of example only, with reference to embodiments thereof which are described with the aid of the accompanying drawings, wherein:

Figure 1 schematically illustrates a networked system for real-time audio/visual services comprising a plurality of client host and a server;
Figure 2 schematically illustrates a client host comprising a dejittering buffer and a software PLL-like architecture, wherein dejittering and clock recovering is achieved in accordance with an embodiment of the present invention; and
Figure 3 diagrammatically illustrates the mechanism of timing synchronization via monitoring the position movement of the dejittering buffer.

with the reference position (defined as the buffer half-size position) for a given period T (for example, every minute). If the buffer position is moving in the direction of emptiness, the counter TC' should be upwardly adjusted by adding an offset. If the buffer position is moving in the other direction (fullness), the TC' value should be downwardly adjusted. The drift rate r of clock unsynchronization between server and client can be determined by:

$$r = (p_2 - p_1)/T$$

[0023] It should be noted that the buffer position mentioned above is in terms of packet index offset rather than the byte number offset.

[0024] It is also possible to implement the present technique with the use of one buffer only (no additional synchronizing buffer available). In such case, data packets are fed into the A/V decoder directly from the dejittering buffer at the scheduled time. The above clock drift rate r will include the component of network jitter J and should be eliminated before the TC' is adjusted. The interarrival jitter J can be derived from the two sequenced RTP packets. The jitter J is defined to be the mean deviation of the difference (D) in packet spacing at the receiver compared to the sender for a pair of packets. For example, if T_{sa} is the RTP timestamp from packet a, and T_{rb} is the time of arrival in RTP timestamp units for packet b, then for these two packets, we have,

$$j = D(a,b)$$

$$= (T_{rb} - T_{ra}) - (T_{sb} - T_{sa})$$

$$= (T_{rb} - T_{sb}) - (T_{ra} - T_{sa})$$

[0025] The j is calculated continuously as each data packet is received from server, then according to the formula, we have,

$$J = J + (j - J)/16$$

[0026] This algorithm provides an optimal first-order estimator and the gain parameter 1/16 gives a good noise reduction ratio while maintaining a reasonable rate of convergence (see RTP specification and related references).

[0027] Unlike the traditional method (using a PLL circuitry) and other available technique (such as that disclosed in European patent No. EP779725, entitled "Method and Apparatus for Delivering Simultaneous Constant Bit Rate Compressed Video Streams At Arbitrary Bit Rates with Constrained Drift and Jitter", which uses two levels of synchronization named coarse-grain and fine-grain for video streams to control drift and jitter), the present method is a more independent and less constrained client-based approach. It provides the network adaptability via a simple software solution (can be implemented in hardware as well) for an end system and can be applied in a wide range of network applications. The effect of the present technique includes:

- increasing the adaptability of decoding systems (in terms of dejittering and clock synchronization);
- simplifying the decoder implementation (no PLL circuitry required);
- handling more types of audio/visual streams (not only MPEG2 system streams);
- application to different types of environments (unicast as well as multicast);
- beneficial to the effort of expanding real-time A/V services over IP-based network such as Internet.

[0028] The foregoing detailed description of embodiments and implementations of the invention have been presented by way of example only, and is not intended to be considered limiting to the present invention as defined in the claims appended hereto. Numerous alternative embodiments may be devised by those skilled in the art without departing from the spirit and scope of the invention.

[0029] Throughout this specification and the claims which follow, unless the context requires otherwise, the word "comprise", and variations such as "comprises" and "comprising", will be understood to imply the inclusion of a stated integer or step or group of integers or steps but not the exclusion of any other integer or step or group of integers or steps.

Claims

1. A method for clock variation compensation in a real-time audio/visual system in which encoded A/V data in a plurality of data packets are delivered to at least one client over a network from a server at a substantially constant bit rate, said plurality of data packets including data packets containing timestamp data, the method comprising the steps of:

   receiving and buffering the data packets in a first buffer at a client;
   passing selected data packets from the first buffer to a second buffer at scheduled times based on a comparison between a local clock of the client and timestamp data corresponding to the selected data packets; and
   passing the data in the second buffer to a data decoder of the client.

2. A method as claimed in claim 1, including monitoring the fullness of the second buffer to derive a drift rate, and adjusting the client local clock based on

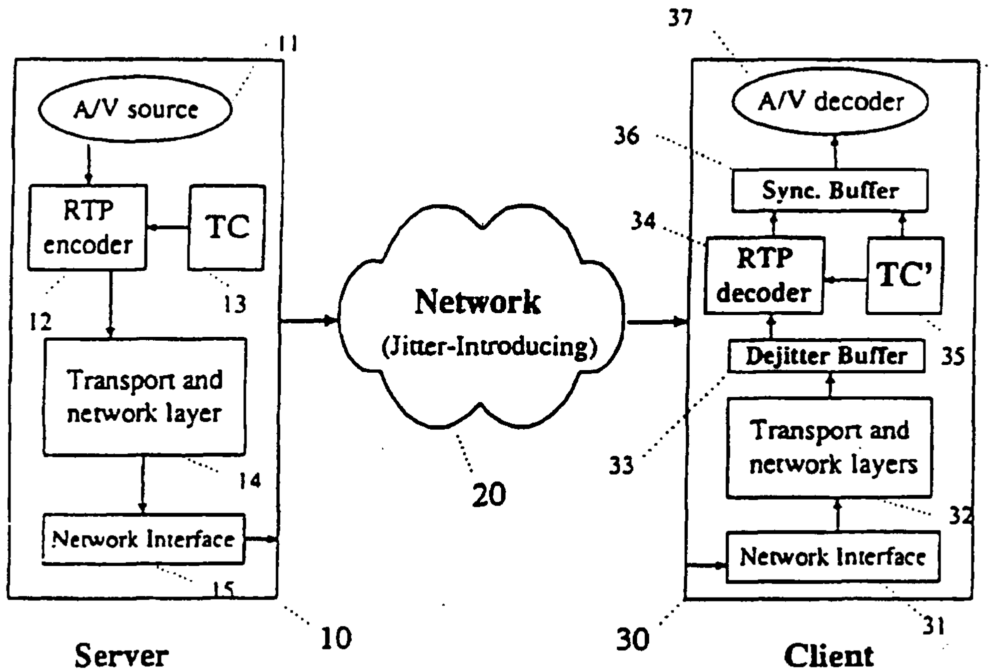
A/V source
RTP encoder
TC
Transport and network layer
Network Interface
11
12
13
14
15
10
Server
Network
(Jitter-Introducing)
20
A/V decoder
Sync. Buffer
RTP decoder
TC'
Dejitter Buffer
Transport and network layers
Network Interface
37
36
34
35
33
32
31
30
Client

Figure 1

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 987 894 A3

(12) EUROPEAN PATENT APPLICATION

(88) Date of publication A3:
19.04.2000 Bulletin 2000/16

(51) Int Cl.7: **H04N 7/24**, H04L 12/64

(43) Date of publication A2:
22.03.2000 Bulletin 2000/12

(21) Application number: **99304426.2**

(22) Date of filing: **08.06.1999**

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(30) Priority: **07.09.1998 SG 9803578**

(71) Applicant: **VICTOR COMPANY OF JAPAN, LTD.**
Yokohama-Shi, Kanagawa-Ken (JP)

(72) Inventor: **XU, Jinghong**
Singapore 760247 (SG)

(74) Representative: **Crawford, Andrew Birkby et al**
A.A. Thornton & Co.
235 High Holborn
London WC1V 7LE (GB)

(54) **A dejittering and clock recovery technique for real-time audio/visual network applications**

(57) In a real-time audio/visual system in which A/V data is conveyed over a jitter-introducing network, dejittering and clock recovery processes can be achieved without requiring a Phase Locked Loop (PLL). At the server, audio/video streams are encoded into transport packets before being sent out. At the client, the dejittering process is achieved by a dejittering buffer using the embedded timestamps in the transport packets and a client decoding clock. The delay variations of data arriving are removed after the client buffering process. At the scheduled time, each data packet is shifted to a synchronizing buffer and then fed to the A/V decoder according to the speed of A/V stream. The clock synchronization between client and server is achieved by a synchronizing buffer whose half-size position is taken as the reference. By monitoring the movement of the buffer fill position over a given period, the drift rate of clock unsynchronization between client and server can be derived and, therefore, the client's clock can be adjusted to synchronize with the server's clock based on the derived drift.

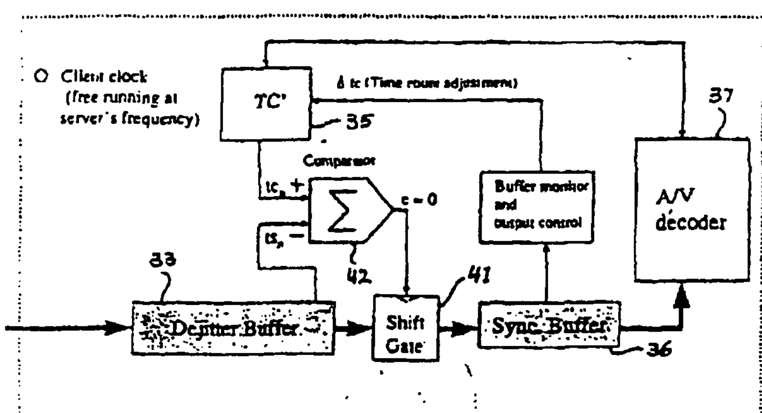

Figure 2

Printed by Jouve, 75001 PARIS (FR)

BNSDOCID: <EP________0987894A3_I_>